application discloses the correct form of the equation, albeit with a different choice of letters as variables. The correlation between the formula found on page 12, line 18 of the parent application and the corrected formula in this application will now be explained.

In the parent application, the subscript "c" as applied to the variables "u" and "G" in the parent application corresponds with the subscript "i" as applied to the variables "u" and "G" in the present application. Thus, the variables u_c and G_c found in the parent application correspond with the variables u_i and G_i found in the present application.

The parent application discloses the following relationship between u_c and G_c in the family of merging gain functions:

$$u_c = u_0 \left(\frac{G_0}{G_c} \right)^{\frac{p}{1-p}}$$

Substituting the values u_i and G_i for u_c and G_c respectively, and by setting G_0 (which represents the gain of a healthy cochlea) to 1, one is left with the following equation:

$$u_i = u_0 \left(\frac{1}{G_i} \right)^{\frac{p}{1-p}}$$

Further, recognizing the rule that:

$$\left(\frac{1}{x} \right)^q = x^{-q}$$

the following equation results:

$$u_i = u_0 G_i^{\frac{-p}{1-p}}$$

which matches the amendment identified in paragraph 2. Therefore, in correcting the formula on page 27, line 3 of the present application, Applicant has not entered new matter and has instead amended the

application to correctly recite subject matter already disclosed in the parent application.

The specification amendment identified in paragraph (3) is made to simply correct typographical errors in presenting the formula expressing the relationship between G_a and G_i. The corrected version of the formula which appears in this amendment is not new matter as it is the result of the application of algebra to the stated conditions.

Starting on page 26, line 12, the application notes that the "transfer functions of the present invention are linear for small signals and sign-preserving power-law compressive for larger signals." The transfer function $f(u,u_0,p)$ is disclosed as:

$$f(u,u_0,p)=u_0\,\mathrm{sgn}(u_0)\left[\left(1+\left(\frac{u}{u_0}\right)^{2n}\right)^{p}-1\right]^{\frac{1}{2n}}$$

wherein p represents the compression power, u represents the instantaneous input amplitude, u_0 represents a normalization coefficient, and n represents a smoothness coefficient. A family of merging transfer functions is obtained from $f(uG_i,u_i,p)$ wherein the instantaneous input amplitude u_i is amplified by G_i.

As described on page 27, such a transfer function can be implemented with pre-amplification G_a and post-amplification G_b. As one of ordinary skill in the art would readily recognize, the resultant condition is as follows:

$$G_b f(uG_a,u_0,p)=f(uG_i,u_i,p)$$

Using the transfer function's large signal and small signal responses, the correct version of the formula appearing on page 27, line 13 is easily obtained. For small signals, the response is linear and the formula reduces to:

$$G_aG_b=G_i$$

From the large signal response, the following equation is obtained:

$$G_b u_0 \left(\frac{G_a}{u_0}\right)^p = u_i \left(\frac{G_i}{u_i}\right)^p$$

Substituting for G_b with G_i/G_a, the resultant equation is:

$$\frac{G_i}{G_a} u_0 \left(\frac{G_a}{u_0}\right)^p = u_i \left(\frac{G_i}{u_i}\right)^p$$

which further reduces to:

$$G_a^{\,p-1} u_0^{\,1-p} = u_i^{\,1-p} G_i^{\,p-1}$$

The further application of algebra results in:

$$\left(\frac{u_o}{u_i}\right)^{1-p} = \left(\frac{G_i}{G_a}\right)^{p-1} = \left(\frac{u_i}{u_0}\right)^{p-1}$$

which further reduces to:

$$\frac{u_i}{u_0} = \frac{G_i}{G_a}$$

Thereafter, by substituting u_i/u_o with $G_i^{-p/(1-p)}$ from the corrected version of the equation on page 27, line 3, one is left with the formula:

$$G_i^{\frac{-p}{1-p}} = \frac{G_i}{G_a}$$

which reduces to:

$$G_i^{-1} G_i^{\frac{-p}{1-p}} = \frac{1}{G_a} = G_i^{\frac{-1}{1-p}} = \frac{1}{G_a}$$

which further reduces to:

$$G_a = G_i^{\frac{1}{1-p}}$$

Thus, it can be seen that the corrected version of the formula for G_a appearing on page 27, line 13 is not new matter as it is simply the algebraic extension of the conditions disclosed in the application on page 26, line 12 through page 27, line 13.

The specification amendments identified in paragraphs (4) and (5) are to correct obvious typographical errors wherein "sent" was typed rather than "seen" and "signal" was typed rather than "single" respectively.

The specification amendment identified in paragraph 6 is made to correct a transposition of digits in the numerator and denominator of the exponential term of the formula.

The specification amendment identified in paragraph (7) is made to correctly list "296" as the numeral referencing the "channel" as shown in Figure 21.

The specification amendment identified in paragraph (8) is an elimination of the reference numerals referring to bandpass filters that had already previously been defined.

The specification amendment identified in paragraph (9) is an amendment to correct an unneeded use of the word "the".

The specification amendments identified in paragraphs (10) and (11) are made to correct typographical errors where two "less than or equal to" symbols were incorrectly identified as "less than" symbols. Support for the amendment can be found in Figure 22 which clearly identifies that "less than or equal to" symbols should have been used.

which ~~easily~~ further reduces to:

$$G_a = G_i^{\frac{1}{1-p}}$$

Thus, it can be seen that the corrected version of the formula for G_a appearing on page 27, line 13 is not new matter as it is simply the algebraic extension of the conditions disclosed in the application on page 26, line 12 through page 27, line 13.

The specification amendments identified in paragraphs (4) and (5) are to correct obvious typographical errors wherein "sent" was typed rather than "seen" and "signal" was typed rather than "single" respectively.

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The specification amendments identified in paragraphs (10) and (11) are made to correct typographical errors where two "less than or equal to" symbols were incorrectly identified as "less than" symbols. Support for the amendment can be found in Figure 22 which clearly identifies that "less than or equal to" symbols should have been used.

The specification amendment identified in paragraph (12) is made to correct the accidental omission of the word "ear". The fact that the sentence should read "outer ear caused" rather than "outer caused" is obvious from the context of the paragraph.

The specification amendment identified in paragraph (12) is made to correct the accidental omission of the word "ear". The fact that the sentence should read "outer ear caused" rather than "outer caused" is obvious from the context of the paragraph.

Further, the amendments to the claims are made to correct inadvertent typographical and clerical errors (correcting claim 4 to read "at least one channel" instead of "at least channel", correcting claims 40, 42, and 44 to read "the magnitude of a quiescent gain" instead of "the magnitude of said linear gain", correcting claim 42 to read "1/p" instead of "$1/p_1$", and correcting claim 44 to read "defined by the general formula" instead of "defined the general formula". All changes are fully supported by the specification.

Favorable examination is respectfully requested.

Respectfully submitted,

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MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 15, LINES 30-34

(additions bolded; deletions bolded and bracketed)

Fig. 13 is the first basic [**IWRDC**] **IWDRC** transducer in Fig. 12, with an adaptive compression threshold and providing the specified maximum low-level linear gain correction and intermediate-level power-law compression;

MARKED-UP COPY OF SPECIFICATION PARAGRAPH EXTENDING FROM PAGE 26, LINE 27 TO PAGE 27, LINE 3

(additions bolded; deletions bolded and bracketed)

A family of merging transfer functions, in accordance with the present invention, is obtained from $f(uG_i, u_i, p)$, wherein the instantaneous input amplitude, u, is amplified by G_i, and by requiring the following relationship between G_i and u_i:

[$u_i = u_o G_i^{\frac{p}{1-p}}$ **]**

$$u_i = u_o G_i^{\frac{-p}{1-p}}$$

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 27, LINES 7-25

(additions bolded; deletions bolded and bracketed)

An alternative algorithmic implementation of the family of merging transducer functions is also provided, which is preferred when the basic transducer is realized with an optimized analog or digital module. The module maintains a fixed transducer function and uses pre- and post-amplification G_a and G_b that depend upon G_i:

$$[\ G_1 = G_a^{\frac{1}{1-p}}, \quad G_a G_b = G_i\ ]$$

$$G_a = G_i^{\frac{1}{1-p}}, \quad G_a G_b = G_i$$

Empirical constraints have been discovered for the smoothness of the transition from linear response to compressive response. Cochlear response functions for simple tone signals are well represented with the choice of the smoothness parameter, n, being set to 2. Pilot psychophysical study of speech intelligibility for normal-hearing listeners with the amplifier described in Fig. 1 have demonstrated significantly better performance (80% vs. 60%) when a smooth transition is provided between the linear region and compressive region (n=2), as opposed to a sharp transition (a sharp transition being a transition having a discontinuous derivative) wherein "n" is large ($n \to \infty$).

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 29, LINES 6-28

(additions bolded; deletions bolded and bracketed)

Fig. 5 shows the required nonlinear gain corrections, for both the mildly impaired cochlea and the moderately impaired cochlea of Fig. 4. The gain correction required for the mildly impaired cochlea is represented by curve 208 while the gain correction required for the moderately impaired cochlea is represented by curve 210. These curves are derived from Fig. 4 by noting the horizontal distance in dB between the responses of the healthy and the impaired cochleae at the signal levels in dB shown. For example, at 20 dB SPL in Fig. 4, curve 200, representing the response of a healthy cochlea, shows a displacement of about 2.5 nanometers. A gain of slightly less than 40 dB is required to provide the same displacement for the moderately impaired cochlea, while a gain of only 20 dB is required for the mildly impaired cochlea. At 40 dB SPL, a gain of slightly less than 30 dB is required for the moderately impaired cochlea, while a gain of 20 dB still suffices for the mildly impaired cochlea. At about 60 dB SPL, the gain required for both the mildly and the moderately impaired cochlea is about 20 dB. As can be [**sent**] **seen** at greater SPLs, the required gain is essentially the same for both the mildly and moderately impaired cochlea, and this gain diminishes as SPL increases, approaching 0 dB for levels above approximately 100 dB SPL.

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 33, LINES 13-22

(additions bolded; deletions bolded and bracketed)

It should be noted that while the effects of shifting compression thresholds shown in Figs. 7-9 are general, the detailed calibration of the abscissa is dependent upon the division of the audio spectrum into separate bands in the amplifier design. In the six-channel amplifier that was investigated, most of the signal energy was divided among the lowest four octave bands. Therefore, to achieve similar results, a [**signal**] **single** channel wideband adaptive nonlinear amplifier would require over 6 dB greater shifts in compression threshold relative to the input signal.

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 35, LINES 3-15

(additions bolded; deletions bolded and bracketed)

This relationship is shown in Fig. 11, both graphically and analytically. The slightly modified gain specification G'(u) shown in Fig. 10 by the dotted lines is obtained by multiplying G(U) by the reciprocal of the describing-function factor, D(p):

$$D(p)=\left(\frac{2}{\sqrt{\Pi}}\right)\frac{\Gamma(1+0.5p)}{\Gamma(1.5+0.5p)2^{0.5(1-p)}};$$

wherein $\Gamma(\ )$ is the gamma function

This modification is further defined as equivalent to shifting the nonlinear thresholds to slightly higher values, as follows:

[$U_1' = U_1 D(1/3)^{-2/3}$, $U_2' = U_2 D(1/3)^{-2/3}$, $U_3' = U_3 D(1/4)^{-4/3}$]

$U_1' = U_1 D(1/3)^{-3/2}$, $U_2' = U_2 D(1/3)^{-3/2}$, $U_3' = U_3 D(1/4)^{-4/3}$

MARKED-UP COPY OF SPECIFICATION PARAGRAPH EXTENDING FROM PAGE 40, LINE 22 TO PAGE 41, LINE 7

(additions bolded; deletions bolded and bracketed)

The data set provided to the next lower channel 296 (2-4 kHz being the preferred range) is obtained by passing $S_i[n]$ through filter 292 which eliminates the frequencies in the highest range (the frequency range of channel 294) by means of lowpass digital filtering, and then downsamples the filtered data set by eliminating every second sample. Downsampled signal 298 is processed by the nonlinear transducer 108 of channel [**298**] **296**, with pre- and post-bandpass filtering (BPFs 104 and 112). Then, the signal leaving channel 296 is upsampled to the original sampling rate by lowpass and interpolation filter 302 and added to the output of the other channels. The upsampling is accomplished by inserting a zero amplitude sample after every second sample in the output of the second filter of the channel, and then interpolating the inserted sample amplitudes using lowpass filtering with amplitude scaling of 2. This scheme is repeated successively for each lower octave channel. The outputs of each channel are summed through the interpolation filters 302 and adders 304 to generate the resultant amplified sound signal Sum[n] provided to DAC 306.

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 41, LINES 8-24

(additions bolded; deletions bolded and bracketed)

In addition to the savings in data processing, the multirate design allows use of identical bandpass and lowpass filters [**104 and 112**] in all channels. Many conventional techniques are available for filter design. A preferred design uses 21 tap FIR bandpass filters (with cutoff frequencies $\pi/4$ and $\pi/2$, and 22 tap lowpass filters (with cutoff frequency 0.30π), each synthesized as a windowed Butterworth IIR filter. The equalization stages 298 shown in Fig. 21 for the upper channels (all but the lowest frequency channel 300) are delays added to provide equal average group delay for the signals processed in each channel. Thus, a broadband audio signal with a well defined temporal epoch will be compactly reconstructed in a similar, but delayed, time window in the multichannel output Sum[n]. In an alternative design with a single channel broadband compressive audio amplifier, a linear phase filter design is preferred.

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 44, LINES 1-9

(additions bolded; deletions bolded and bracketed)

Therefore, it is desirable that the compression threshold not be reduced until the signal's average sound level drops by a triggering amount, which identifies [**the**] when a need exists to drop the compression threshold to adapt to a quieter environment. However, it is also desirable that the compression threshold quickly track increases in the sound signal's average sound level to minimize overamplification of background noise. The flowchart describes how these goals are accomplished.

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 44, LINES 21-32

(additions bolded; deletions bolded and bracketed)

At step 1004, controller determines the peak (maximum magnitude) X_i of signal block $S_i[n]$. X_i will be the sample of $S_i[n]$ having the largest amplitude. Next, at step 1006, the controller will sort X_i with respect to the currently stored value of X. Essentially, the controller will determine (1) whether the peak is increasing from the stored peak (is $X_i > X$?), (2) whether the peak is decreasing an insignificant amount [**(is $\rho X < X_i < X$?)**] **(is $\rho X < X_i \leq X$?)**, and (3) whether the peak is decreasing a significant amount, that is, decreasing by a triggering amount [**(is $X_i < \rho X$?)**] **(is $X_i \leq \rho X$?)**. The parameter ρ is used to control the triggering amount. Preferably, $0<\rho<1$, and more preferably ρ is set equal to ½.

MARKED-UP COPY OF SPECIFICATION PARAGRAPH ON PAGE 50, LINES 7-23

(additions bolded; deletions bolded and bracketed)

It is preferred to choose a common compression ratio (1/p) for all of the channels, so that the quiescent transducer responses for the different channels merge at high signal levels, while differing at low levels only in the compression threshold determined by G_1. The smallest compression ratio should be chosen from among the values 2, 3 and 4, to provide the range compression needed for signal frequencies of 0.5, 1.0 and 2.0 kHz. These frequencies are found to be most important for speech communication. Greater hearing losses at other frequencies should be corrected only to the extent possible with the chosen compression ratio. Compensation should be included for the loss of normal free-field acoustic amplification by the outer **ear** caused by use of standard earmolds or insert earphones. A preferred compensation provides a constant 14 dB gain emphasis for the 2-4 kHz channel relative to the other channels.

MARKED-UP COPY OF AMENDED CLAIMS 4, 40, 42, 44

(additions bolded; deletions bolded and bracketed)

4. (amended) The hearing amplification device of claim 3 wherein said at least one channel is configured to have its compression threshold initially set to a predetermined quiescent level, and wherein said at least **one** channel is further configured to adjust said compression threshold such that said compression threshold is in a range of about said predetermined quiescent level to about 20 decibels below an average sound level of at least a portion of said sound signal.

40. (amended) The device of claim 35 wherein an asymptotic representation of said transfer function TA1 is defined by the general formula:

$$TA=TA(u,A,U,p),$$

wherein for $|u| < U$:

$$TA(u,A,U,p)=Au$$

wherein for $|u| > U$

$$TA(u,A,U,p)=\operatorname{sgn}(u)AU\left|\frac{u}{U}\right|^{p}$$

wherein:

$$TA1=TA1(u,U_C)=TA(u,A(U_C),U_C(Y),p);$$

wherein $U_C(Y)= U_1$ for Y less than U_1 and $U_C(Y) = Y$ for Y greater than or equal to U_1, wherein U_1 represents a quiescent level for said compression threshold, wherein U_C represents an adjusted compression threshold, wherein Y represents a control signal from said controller for controlling said compression threshold, wherein u represents said transducer input, wherein p represents a compression power, and wherein A represents a magnitude of gain, wherein for Y less than U_1:

$$A=G_1$$

and wherein for Y greater than or equal to U_1:

$$A = G_1 \left| \frac{U_1}{U_C} \right|^{1-p}$$

wherein G_1 represents the magnitude of [**said linear**] **a quiescent** gain.

42. (amended) The device of claim 41 wherein an asymptotic representation of said transfer function is defined as a cascade of two functions TA1 and TA2,wherein both TA1 and TA2 are defined **by** the general formula:

TA=TA(u,A,U,p),

wherein for |u| < U:

$$TA(u, A, U, p) = Au$$

wherein for |u| > U

$$TA(u, A, U, p) = \operatorname{sgn}(u) AU \left| \frac{u}{U} \right|^p$$

wherein:

TA1=TA1(u,U_C)=TA(u,A(U_C),U_C(Y),p);

wherein U_C(Y)= U_1 for Y less than U_1, U_C(Y) = Y for Y greater than or equal to U_1 and less than or equal to U_2, and U_C(Y)=U_2 for Y greater than U_2, wherein U_1 represents a quiescent level for said compression threshold, wherein U_2 represents said decompression threshold, wherein U_C represents an adjusted compression threshold, wherein Y represents a control signal from said controller for controlling said compression threshold, wherein u represents said transducer input, wherein p represents a compression power, and wherein A represents a magnitude of gain, wherein for Y less than U_1:

$$A = G_1$$

and wherein for Y greater than or equal to U_1:

$$A = G_1 \left| \frac{U_1}{U_C} \right|^{1-p}$$

wherein G_1 represents the magnitude of [**said linear**] **a quiescent** gain; and

wherein for $TA2=TA2(u)=TA(u,1,U_2,p_2)$, wherein u represents TA1, wherein U_2 represents said decompression threshold, and wherein p_2 represents [$1/p_1$] **1/p**.

44. (amended) The device of claim 43 wherein an asymptotic representation of said transfer function is defined as a cascade of three functions TA1, TA2, and TA3, wherein TA1 , TA2, and TA3 are each defined **by** the general formula:

$$TA=TA(u,A,U,p),$$

wherein for $|u| < U$:

$$TA(u,A,U,p)=Au$$

wherein for $|u| > U$

$$TA(u,A,U,p)=\operatorname{sgn}(u)AU\left|\frac{u}{U}\right|^{p}$$

wherein:

$$TA1=TA1(u,U_C)=TA(u,A(U_C),U_C(Y),p_1);$$

wherein $U_C(Y)= U_1$ for Y less than U_1, $U_C(Y) = Y$ for Y greater than or equal to U_1 and less than or equal to U_2, and $U_C(Y)=U_2$ for Y greater than U_2, wherein U_1 represents a quiescent level for said compression threshold, wherein U_2 represents said decompression threshold, wherein U_C represents an adjusted compression threshold, wherein Y represents a control signal from said controller for controlling said compression threshold, wherein u represents said transducer input, wherein p_1 represents a first compression power, and wherein A represents a magnitude of gain, wherein for Y less than U_1:

$$A=G_1$$

and wherein for Y greater than or equal to U_1:

$$A=G_1\left|\frac{U_1}{U_C}\right|^{1-p}$$

wherein G_1 represents the magnitude of [**said linear**] **a quiescent** gain;

wherein for $TA2=TA2(u)=TA(u,1,U_2,p_2)$, wherein u represents TA1 or TA3, wherein U_2 represents said decompression threshold, and wherein p_2 represents $1/p_1$; and

wherein for $TA3=TA3(u)=TA(u,1,U_3,p_3)$, u represents TA1 or TA2, wherein U_3 represents said attenuation threshold, and wherein p_3 represents a second compression power.